citation is wrong. $\delta(j)$ is defined on page 26, line 18. The Examiner takes the position that the time shifts are used only in minimizing distortions of unvoiced signals and, therefore, an argument that time shifting is employed for minimizing distortions in both voiced and unvoiced signals is outside the scope of the disclosed invention. First of all, the disclosure stands on its own, and any arguments, however interpreted by the Examiner, are not part of the disclosure. So from that point of view, the rejection is improper. Second, the Examiner has quite clearly not understood the invention or the relevance of the equations she has cited as supporting her position that there is no time shifting in voiced mode. In the paragraphs preceding equation 11 on page 24, Applicant explains that an amplitude codebook or a polarity codebook can be used. The case of a polarity codebook is described wherein codebook 351 is used for voiced sound and codebook 352 is used for unvoiced sound. The point here is that Applicant has used separate codebooks for voiced and unvoiced speech signals so as to minimize calculations without adversely affecting sound quality in terms of background noise.

Since the claims are clearly supported by the specification and drawings as originally filed, the rejection under 35 U.S.C. §112, first paragraph, is without ground and should be withdrawn.

Claims 1 to 11 were additionally rejected under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,704,003 to Kleijn et al. in view of U.S. Patent No. 5,751,903 to Swaminathan et al. and U.S. Patent No. 5,657,418 to Gershon et al. This rejection is respectfully traversed for the reason that the combination of references relied on by the Examiner fails to show or suggest the claimed invention.

The claimed invention is directed to a speech coding apparatus for coding a speech signal at a low bit rate with high quality. The invention effectively suppresses deterioration in sound quality in terms of background noise while minimizing the calculations required. The speech coding apparatus uses a mode discrimination circuit 370 (Fig. 1) which discriminates the mode on the basis of

the past quantized gain of an adaptive codebook. The past quantized gain is shown in Figure 1 by the input to the mode discrimination circuit 370 from gain quantization circuit 366, and the adaptive codebook 500 also receives an input from gain quantization circuit 366. When a predetermined mode is discriminated as either voiced or unvoiced, a sound source quantization circuit 350 searches combinations of code vectors stored in a code book 351 or 352, which is used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. The gain quantization circuit 366 quantizes gains by using a gain codebook 380.

A main characteristic feature of the present invention as claimed in claim 1 is that a speech coding apparatus comprises a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from the discrimination section indicates a predetermined mode, and searches combinations of code vectors stored in the codebook and a plurality of shift amounts used to shift positions of the pulse so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech.

Specifically, claim 1 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". This sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a *past quantized gain* of an adaptive codebook [**500**]" (emphasis added), and "a sound source quantization section [**350**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and *searches combinations of code vectors stored in*

said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech" (emphasis added).

Claim 1 is directed to the first embodiment shown in Figure 1 of the drawings. Claims 2, 3 , 4, and 5 are directed respectively to the second, third, fourth, and fifth embodiments shown in Figures 2, 3, 4, and 5 of the drawings and contain similar limitations.

Specifically, claim 2 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode *on a basis of a past quantized gain of an adaptive codebook*" (emphasis added), and "a sound source quantization section **[350]** which has a codebook **[351, 352]** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and *outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule*" (emphasis added). Note that in the embodiment of Figure 2, a random number generating circuit 600 is added to generate a predetermined number of pulse positions, as described in more detail beginning at page 29, line 23, and continuing to page 30, line 4.

Claim 3 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode *on the basis of a past quantized gain of an adaptive codebook* **[500]**" (emphasis added), and "a sound source quantization section **[350]** which has a codebook **[351, 352]** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing

amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook **[380]** for quantizing gains, and *searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech*" (emphasis added).

Claim 4 recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode *on the basis of a past quantized gain of a adaptive codebook* **[500]**" (emphasis added), and "a sound source quantization section **[350]** which has a codebook **[351, 352]** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on output from said discrimination section, and a gain codebook **[380]** for quantizing gains, and *outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule*" (emphasis added).

Claim 5 is directed to the decoder apparatus and recites, "a mode discrimination section **[530]** for discriminating a voice sound mode and an unvoiced sound mode by *using a past quantized gain in said adaptive codebook* **[520]**" (emphasis added), and "*a sound source signal reconstructing section* **[540]** *for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information based on an output from said discrimination section*, wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section **[560]** which includes spectrum parameters" (emphasis added).

Claims 6 and 7 recite the combination of a speech coding and decoding apparatus and, again, recite limitations similar to those pointed out above. Claim 8 is directed to a speech coding apparatus which comprises, *inter alia*, "sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook *for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech*" (emphasis added). Claims 9, 10 and 11 are dependent on claim 8.

In making the rejection, the Examiner acknowledges that Kleijn et al. do not teach discriminating a voiced/unvoiced mode based on a past quantized gain of an adaptive codebook. The Examiner cited Gerson et al. for a teaching of that a lag parameter, which reflects the periodicity, is used to select a particular coding mode. From this, the Examiner concludes that it would have been obvious to modify Kleijn et al. to implement discriminating a voiced/unvoiced mode based on past quantized gain. What the Examiner is attempting to do is to ignore the clear teaching of the primary reference and attribute to the secondary reference a teaching which is clearly not warranted by the reference. The combination she proposes is based solely on hindsight not permitted by Section 103 of the Patent Statute.

The Examiner further takes the position that Kleijn et al. teach sound source quantization by searching a codebook for code vectors and delays so as to output a combination of code vector and shift amount that minimizes distortion. This is not an accurate characterization of the reference. A characteristic feature of Kleijn et al. is that residual signals are coded by a time shift. That is, as described in column 6, after line 14, the best value for time shift T which can minimize an error output between a signal $r(n-T)$ obtained by shifting the residual signal $r(n)$ by T and a delayed residual signal $r(n-D(n))$ is required, whereby the parameter

required in coding is selected. The Examiner goes on to acknowledge that Kleijn et al. do not teach a multiplexer for the coder or decoder scheme, but cites Swaminathan et al., saying that "it would have been obvious to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as *suggested* by Swaminathan et al." (emphasis added). This, again, is a wholesale reconstruction of the primary reference unwarranted by what the references in fact teach.

The Examiner is reminded of the basic considerations which apply to obviousness rejections as set out in MPEP 2141. Specifically, "When applying 35 U.S.C. 103, the following tenets of patent law must be adhered to:

"(A) The claimed invention must be considered as a whole;

"(B) The references must be considered as a whole and must suggest the desirability and thus the obviousness of making the combination;

"(C) The references must be viewed without the benefit of impermissible hindsight vision afforded by the claimed invention; and

"(D) Reasonable expectation of success is the standard with which obviousness is determined."

The Examiner has taken three rather diverse systems and tried to combine them based on Applicant's own disclosure. It is not even clear that the reconstructions she proposes would result in an operable system, particularly since the references are each based on different principles of operation. The rejection is clearly without merit and should therefore be withdrawn.

In view of the foregoing, it is respectfully requested that the application be reconsidered, that claims 1 to 11 be allowed, and that the application be passed to issue.

Should the Examiner find the application to be other than in condition for allowance, the Examiner is requested to contact the undersigned at the local telephone number listed below to discuss any other changes deemed necessary in a telephonic or personal interview.

A provisional petition is hereby made for any extension of time necessary for the continued pendency during the life of this application. Please charge any fees for such provisional petition and any deficiencies in fees and credit any overpayment of fees to Attorney's Deposit Account No. 50-2041.

Respectfully submitted,

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